Web 2.0 SIP Chatroom solution

Baptiste Bouffaut – June 2009

Introduction

- **This project implements a Web 2.0 to SIP gateway to access SIP services from any web browser:**
  - Chatrooms / Instant Messaging
  - Presence
  - Telephony
  - Multimedia Content Sharing

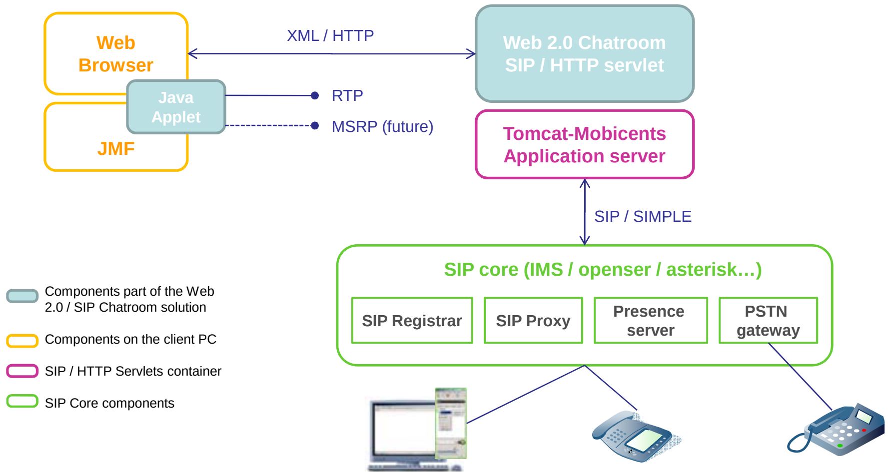
Differentiators

- **Web chatroom based on SIP**: it enables dialoging with any SIP device (softphone, IP phone…)
- **Include telephony services based on SIP**: enables contacting any telephony device from any web browser
- **WEB SIP User-Agent**: does not require any client software installation and maintenance

Technologies

- **Web 2.0 interface**
- **AJAX application**
- **SIP / HTTP servlets** on top of Tomcat-mobicents application server (SIP API 1.1 – JSR 289)
- **Java Applet** based on Java Media Framework (JMF) for media services (telephony, video)

Architecture



Features (1/3)

- **Strong User Authentication** based on session key exchange:
  - The solution keeps a mapping between the user connection characteristics (IP address, browser,…) and the session
  - All XML / HTTP requests are signed with the session key so that they can be authentified by the servlet
- **Identities:**
  - A user can have several private identities (SIP softphone account, WEB account,…) associated to his public identity
  - Public entity is set to “present” as soon as user is registered with at least one of his private identities (he can be connected with several)
- **Presence capabilities:**
  - Possibility to change manually update status
  - Automatic presence update when user is on-call
  - Impossibility to receive incoming call when in DND mode

Features (2/3)

- **Chatroom / Instant messaging capabilities**:
  - User can subscribe to several community chatrooms and access them via the same screen
  - User can start private conversation with any buddy in the chatroom (except those in DND mode)
  - Incoming messages are forked to all the private identities (SIP softphone, WEB account,…) the user is connected with
  - Chatroom historic display

- **Telephony capabilities:**
  - Possibility to have direct audio call with one of the community member
  - Possibility to receive incoming calls from any of the community member (except when in DND mode)
  - Possibility to associate PSTN / PLMN numbers to the public entity
  - Incoming calls are forked to all the private identities and associated numbers the user is connected with

Features (3/3)

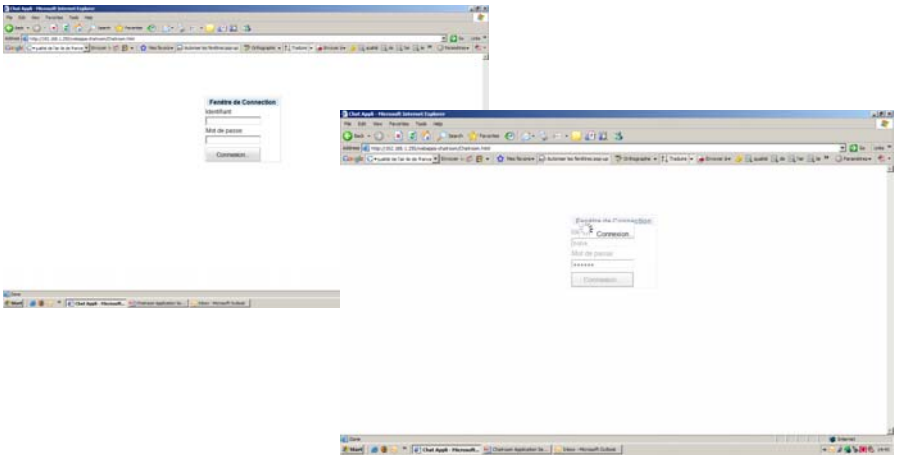
- **Multi-media Content Sharing capabilities**:

  - **Push mode (development on-going):** enables each community member to upload mulitmedia content (photos, audio, videos, document…) on the chatroom so that all members of the chatroom can access it (blog mode)

  - **Peer-to-peer mode (future development):** based on RCS recommendations, use MSRP to transfer multi-media files from peer to peer.

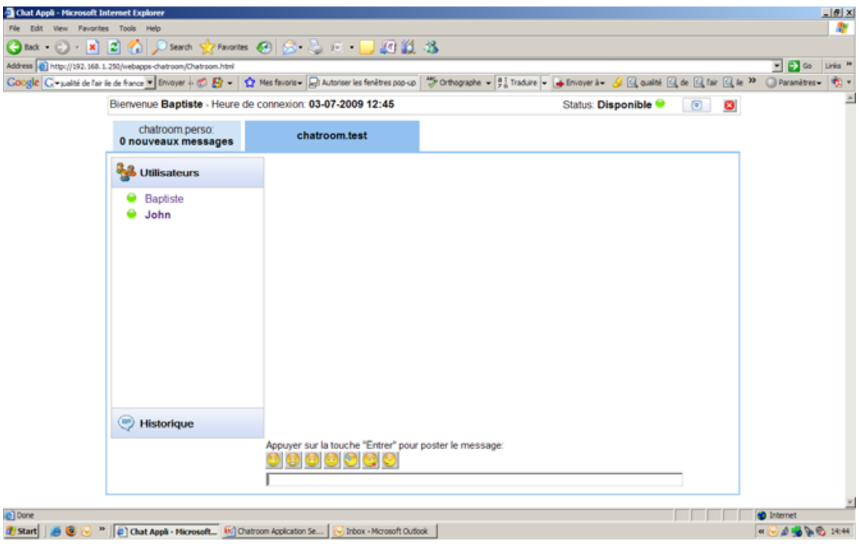
Use-Cases

- **WEB authentication**



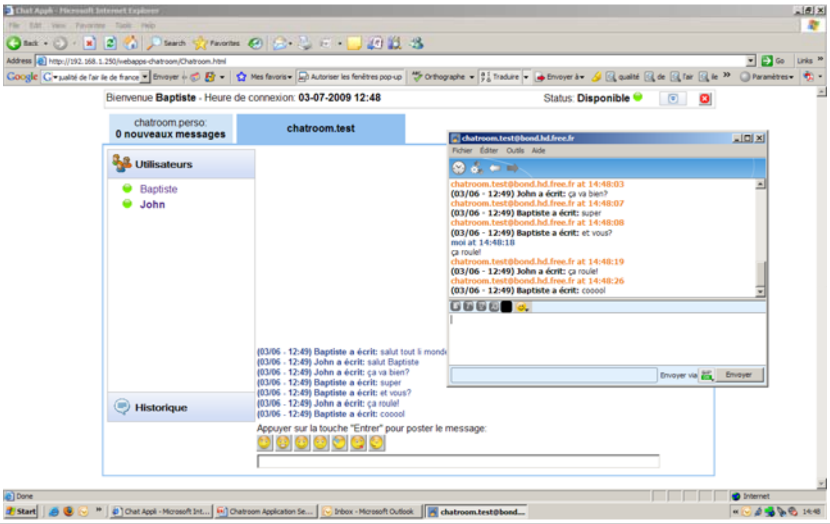
Use-Cases

- **Home page**



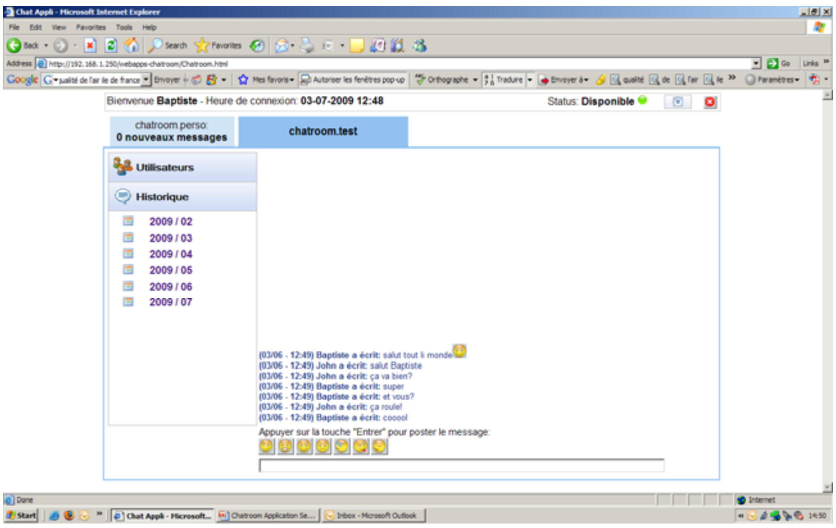
Use-Cases

- **Chat in the chatroom:** Baptiste is connected on his WEB account and John via his SIP softphone. Exchange of messages on the chatroom.



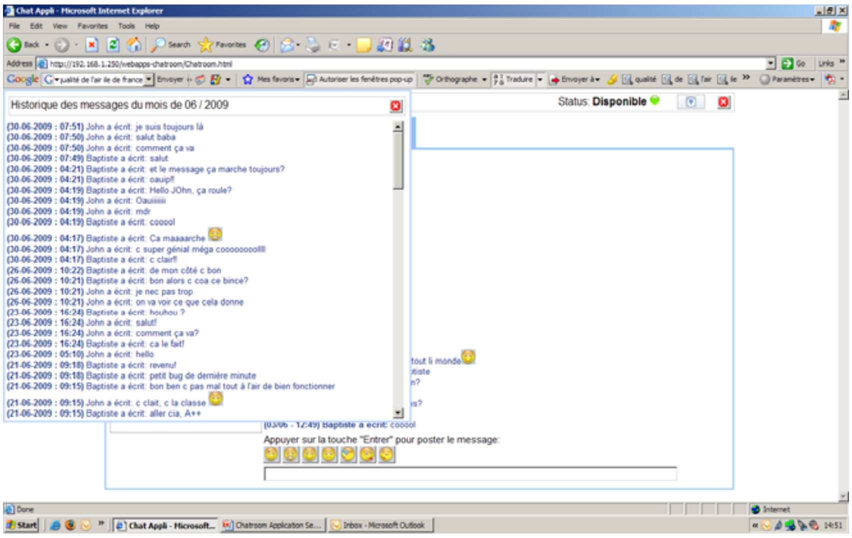
Use-Cases

- **Historic of the chatroom (1 / 2)**



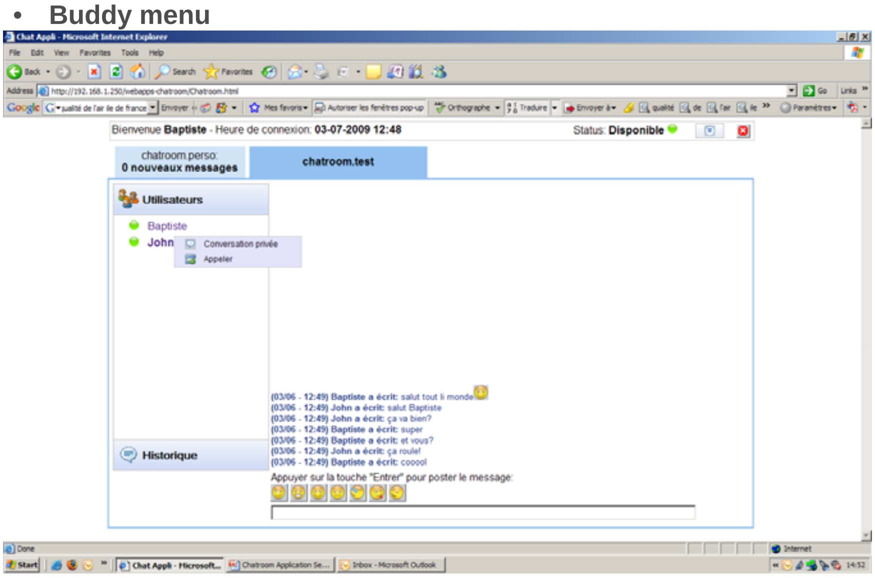
Use-Cases

- **Historic of the chatroom (2 / 2)**



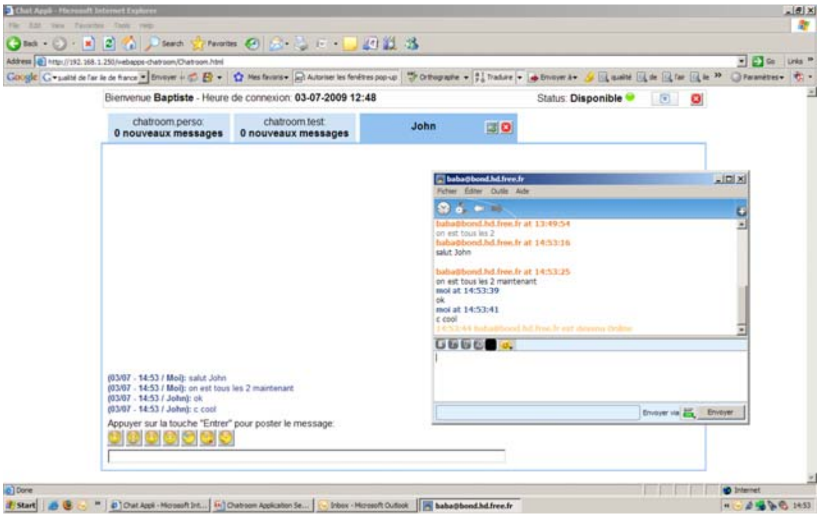
Use-Cases

- **Buddy menu**



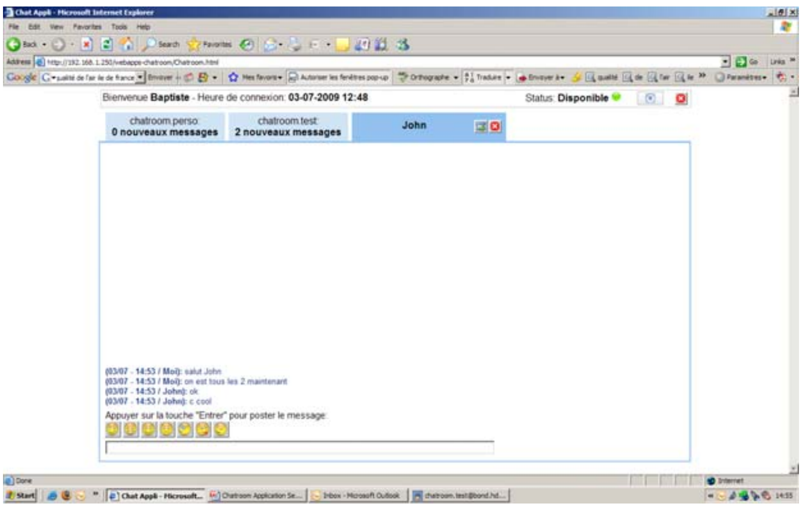
Use-Cases

- **Private chat:** Baptiste is connected on his WEB account and John via his SIP softphone



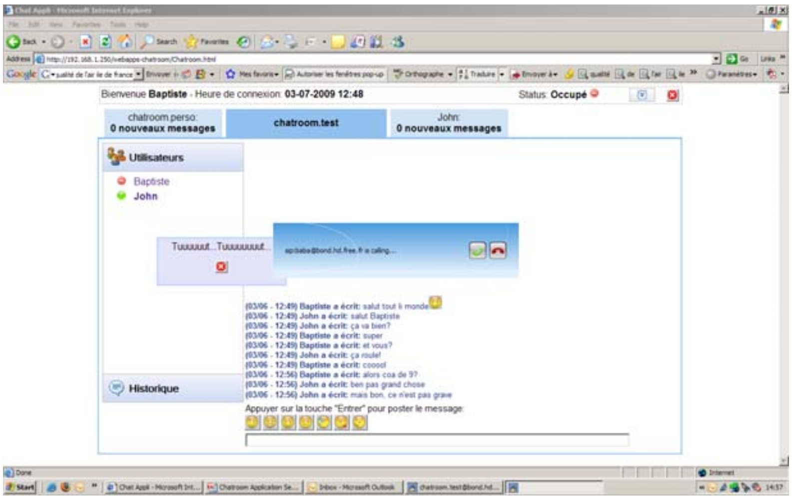
Use-Cases

- **Private chat:** Numbers of new messages are displayed in the top-bar menu for each chatrooms



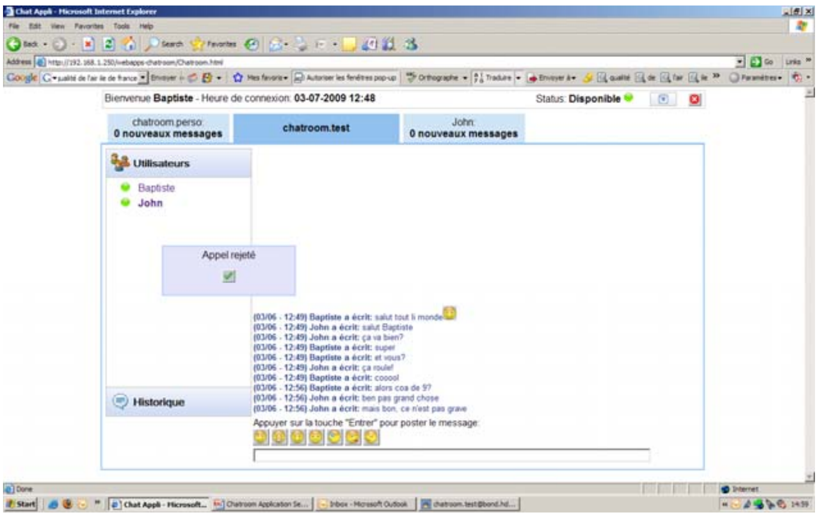
Use-Cases

- **Make a call:** Baptiste is connected on his WEB account and John via his SIP softphone. Baptiste calls John via the buddy menu. John softphone is ringing



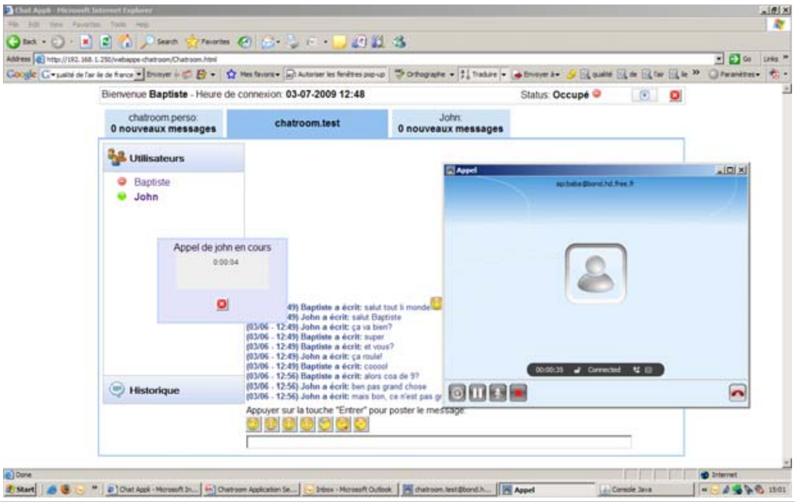
Use-Cases

- **Make a call:** Baptiste is connected on his WEB account and John via his SIP softphone. Baptiste calls John via the buddy menu. John rejects the call.



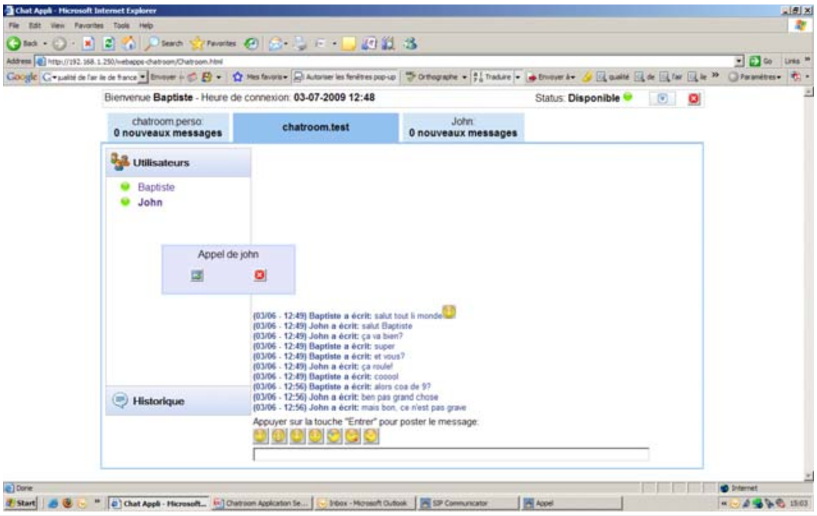
Use-Cases

- **Make a call:** Baptiste is connected on his WEB account and John via his SIP softphone. Baptiste calls John via the buddy menu. John accepts the call on his softphone. Telephony Applet is launched on Baptiste WEB page.



Use-Cases

- **Incoming call:** Baptiste is connected on his WEB account and John via his SIP softphone. John calls Baptiste. Incoming call popup raises on Baptiste WEB page. Baptiste can accept or reject the call. If accepted, Telephony applet is launched.



Use-Cases

- **Presence status update:** Baptiste is connected on his WEB account. He modifies his presence status in the tool-bar menu.